



**VoIP Analog Telephone Adapter**  
**VIP-156**  
**VIP-156PE**  
**User's manual**

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The is a class B device, In a domestic environment, this product may cause radio interference, in which case the user may be required to take adequate measures.

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## Revision

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# Chapter 1 Introduction

## Overview

Based on years of VoIP manufacturing experiences, PLANET Technology VoIP total solutions are known for advanced implementation of standards based telephony with mass deployment capability.

Cost-effective, easy-to-install and simple-to-use, the VIP-156 converts standard telephones to IP-based networks. The service providers and enterprises offer users traditional and enhanced the telephony communication services via the existing broadband connection to the Internet or corporation network.

With the VIP-156, home users and companies are able to save the installation cost and extend their past investments in telephones, conference and speakerphones. The VIP-156 can be the bridge between traditional analog systems and IP network with an extremely affordable investment.

The VIP-156 includes two alternatively Ethernet interface for Internet (PPPoE, DHCP or Fixed IP), or office LAN connection. With adding the auto-provision feature of our IP PBX product - IPX-2000, the VIP-156 can be seamlessly integrated into the telephony network and be used in consumer and business IP telephony service, no expertise required!

The VIP-156 and our IP PBX system integration are the ideal combination for your office daily communications.

## Product Features

- Feature-rich telephone service over home or office Internet/Intranet connection
- Auto-config feature for flexible, ease-of-use IP PBX system integration
- Cost effective, field proven compatibility, and stability
- Web-based and telephone keypad machine configuration
- Remote administrator authentication
- Voice prompt for machine configurations

## VoIP Features

- SIP 2.0 (RFC3261) compliant
- Peer-to-Peer / SIP proxy calls
- Voice codec support: G.711, G.723.1, G.729A/G.729B

- Voice processing: Voice Active Detection, DTMF detection/ generation, G.168 echo cancellation (16mSec.), Comfort noise generation
- In band and out-of-band DTMF support

## Package Content

The contents of your product should contain the following items:

VoIP Telephone Adapter

Power adapter

Quick Installation Guide

User's Manual CD

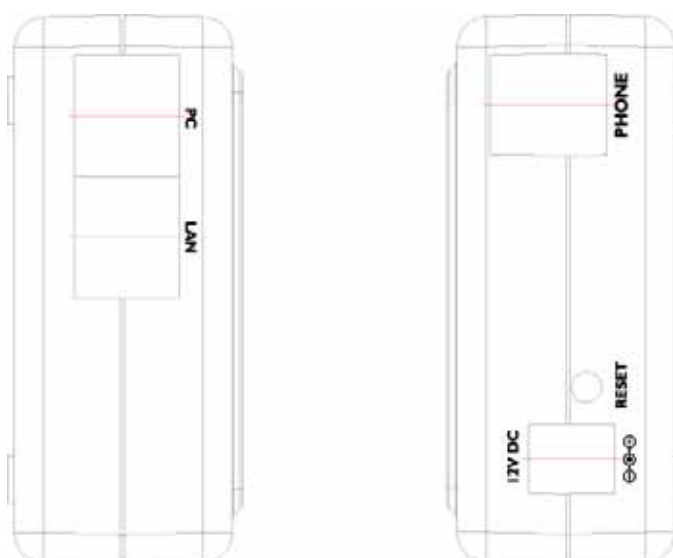
RJ-11 cable x 1

## Physical Details

The following figure illustrates the front/rear panel of VIP-156.



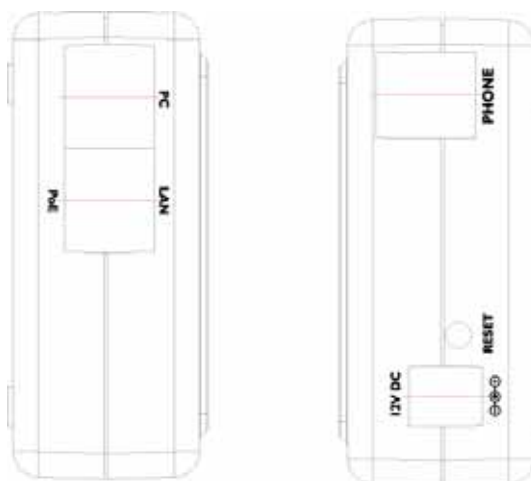
**Front Panel of VIP-156**



**Left / Right Panel of VIP-156**



**Front Panel of VIP-156PE**



Left / Right Panel of VIP-156PE

## LED Display & Button

- |   |               |   |
|---|---------------|---|
| 1 | <b>PC</b>     | RJ-45 connector, to maintain the existing network structure, connected directly to the <b>PC</b> through <b>straight</b> CAT-5 cable  |
| 2 | <b>LAN</b>    | RJ-45 connector, for Internet access, connected directly to <b>Switch/Hub</b> through <b>straight</b> CAT-5 cable.<br>The <b>LAN</b> interface also can be connected with 802.3af PoE switch or converter for power supply ( <b>VIP-156PE</b> ) |
| 3 | <b>12V DC</b> | 12V DC Power input outlet   |
| 4 | <b>Reset</b>  | Reset to the factory default setting  |

### Note

Machine default IP is <http://192.168.0.1>. Press **RESET** button on rear panel over 5 seconds will reset the VoIP Phone Adapter to factory default value. (Except speed dial and call forward settings)

LED Indicators	Descriptions
<b>PWR</b>	Power is supplied to the gateway.
<b>STATUS</b>	The Status LED will be flashing when the machine is operational
<b>LNK/ACT</b>	OFF: the gateway is connected to LAN at 10Mb/s. ON: the gateway is connected to LAN at 100Mb/s.
<b>RING</b>	OFF: the phone is idle. ON: the phone is in use (offhook). Blinking: the phone is ringing.

# Chapter 2

# 2

## Preparations & Installation

### Physical Installation Requirement

This chapter illustrates basic installation of ATA analog Phone Adapter ((“ATA” in the following term))

- Network cables. Use standard 10/100BaseT network (UTP) cables with RJ45 connectors.
- TCP/IP protocol must be installed on all PCs.

For Internet Access, an Internet Access account with an ISP, and either of a DSL or Cable modem

### Administration Interface

---

PLANET ATA provides GUI (Web based, Graphical User Interface) for machine management and administration.

### Web configuration access

To start ATA web configuration, you must have one of these web browsers installed on computer for management

- Netscape Communicator 4.03 or higher
- Microsoft Internet Explorer 4.01 or higher with Java support

Default LAN interface IP address of ATA is **192.168.0.1**. You may now open your web browser, and insert **192.168.0.1** in the address bar of web browser to logon ATA web configuration page.



Enter Network Password

Please type your user name and password  
PLANET Phone Adapter Configuration

User Name

Password

☐ Save this password in your password list

ATA will prompt for logon username/password, please enter: **root / null (no password)** to continue machine administration.

## Note

Please locate your PC in the same network segment (192.168.0.x) of VIP-156. If you're not familiar with TCP/IP, please refer to related chapter on user's manual CD or consult your network administrator for proper network configurations.

## LAN IP address configuration via web configuration interface

Execute your web browser, and insert the IP address (**default: 192.168.0.1**) of VIP in the address bar. After logging on machine with username/password (default: **root / not passwd**), browse to "**Network**" --> "**Network settings**" configuration menu:

### Network Settings

You could configure your network settings in this page.

TCP/IP Configuration	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	<input type="text" value="192.168.0.1"/>
Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="192.168.0.254"/>
DNS Server 1:	<input type="text" value="168.95.1.1"/>
DNS Server 2:	<input type="text" value="168.95.192.1"/>
MAC:	<input type="text" value="00304faabbcc"/>

### Parameter Description

<b>IP address</b>	LAN IP address of ATA  <b>Default:</b> 192.168.0.1
<b>Subnet Mask</b>	LAN mask of ATA  <b>Default:</b> 255.255.255.0
<b>Default Gateway</b>	Gateway of ATA  <b>Default:</b> 192.168.0.254

## Network settings via Keypad commands

The VIP-156 series phone adapters support telephone keypad configurations, please connect analog telephone set and refer to the following table for machine network configurations.

IVR Menu Choice	Machine operation	Parameter(s)	Notes:
#111#	Set DHCP client	None	VIP-156 will change to DHCP Client
#112xxx*xxx*xxx*xxx#	Setup Static IP Address	Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
#113xxx*xxx*xxx*xxx#	Set Network Mask	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#114xxx*xxx*xxx*xxx#	Set Gateway IP Address	Use the * (star) key when entering a decimal point.	Must set Static IP first.
#115xxx*xxx*xxx*xxx#	Set Primary DNS Server	Use the * (star) key when entering a decimal point.	Must set Static IP first.

Following keypad commands can be used to display the network settings enabled on VIP-156 via voice prompt.

IVR Menu Choice	Machine operation	Notes:
#120#	Check IP Address	IVR will announce the current IP address of the VIP-156
#121#	Check network connection Type	IVR will announce if DHCP is enabled or disabled.
#122#	Check the Phone Number	IVR will announce current enabled VoIP number
#123#	Check Network Mask	IVR will announce the current network mask of the VIP-156.
#124#	Check Gateway IP Address	IVR will announce the current gateway IP address of the VIP-156.
#125#	Check Primary DNS Server Setting	IVR will announce the current setting in the Primary DNS field.
#128#	Check Firmware Version	IVR will announce the version of the firmware running on the VIP-156.



Please contact your Internet service provider to obtain the Internet access type, and select the proper network settings in VIP-156 to establish the network connections.

After confirming the modification you've done, Please click on the **Submit** button to apply settings and browse to "**Save & Reboot**" menu to reboot the machine to make the settings effective.

Connection Type	Data required.
<b>Fixed IP</b>	In most circumstances, it is no need to configure the DHCP settings.
<b>DHCP client</b>	The ISP will assign IP Address, and related information.
<b>PPPoE</b>	The ISP will assign PPPoE username / password for Internet access,

**i Hint**

Please consult your ISP personnel to obtain proper PPPoE/IP address related information, and input carefully.  
If Internet connection cannot be established, please check the physical connection or contact the ISP service staff for support information.

### Save Modification to Flash Memory

Most of the VoIP router parameters will take effective after you modify, but it is just temporary stored on RAM only, it will disappear after your reboot or power off the VoIP Phone Adapter, to save the parameters into Flash ROM and let it take effective forever, please remember to press the **Save & Reboot** button after you modify the parameters.

## Save & Reboot

You have to save changes to effect them.

Save Changes:

## Chapter 3

# 3

## Network Service Configurations

### Configuring and monitoring your Phone Adapter from web browser

The ATA integrates a web-based graphical user interface that can cover most configurations and machine status monitoring. Via standard, web browser, you can configure and check machine status from anywhere around the world.

#### Overview on the web interface of ATA

With web graphical user interface, you may have:

- ◆ More comprehensive setting feels than traditional command line interface.
- ◆ Provides user input data fields, check boxes, and for changing machine configuration settings
- ◆ Displays machine running configuration

To start ATA web configuration, you must have one of these web browsers installed on computer for management

- ◆ Netscape Communicator 4.03 or higher
- ◆ Microsoft Internet Explorer 4.01 or higher with Java support

#### Manipulation of ATA via web browser

##### Log on ATA via web browser

After TCP/IP configurations on your PC, you may now open your web browser, and input **http://192.168.0.1** to logon Phone Adapter web configuration page.

Phone Adapter will prompt for logon username/password: **root / null (not password)**



**Enter Network Password**

Please type your user name and password  
PLANET Phone Adapter Configuration

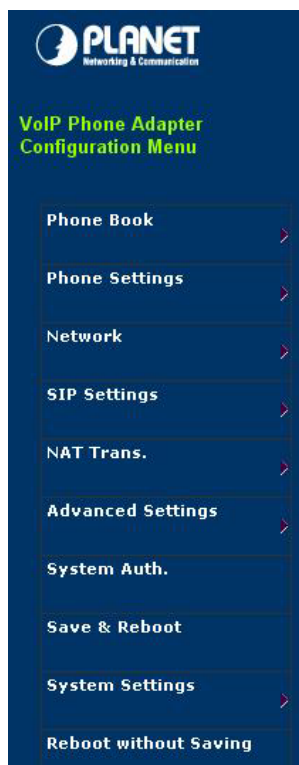
User Name

Password

☐ Save this password in your password list

VIP-156 log in page

When users login the web page, users can see the Phone Adapter system information like firmware version, company...etc in this main page.



## System Information

This page illustrate the system related information.

Company:	PLANET Technology Corp.
Firmware Version:	1.0
Codec Version:	1.0
Contact Address:	11F, No. 96, Min Chuan Road, Hsin Tien, Taipei, Taiwan, R.O.C
Tel:	886-2-22199518
Fax:	886-2-22199528
E-Mail:	<a href="mailto:support_voip@planet.com.tw">support_voip@planet.com.tw</a>
Web Site:	<a href="http://www.planet.com.tw">www.planet.com.tw</a>

*VoIP Phone Adapter main page*

## Chapter 4

# 4

## VoIP Telephone Adapter Configurations

### Speed Dial settings

In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.

If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

If you want to delete all phone numbers, you can click “Delete All” button.

### Speed Dial Phone List

You could set the speed dial phones in this page.

Phone	Name	URL	Select
0			<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected

Delete All

Reset

#### Add New Phone

Position:  (0~9)

Name:

URL:

Add Phone

Reset

## Call forward

This page defines the tones generated to the phone connected to the phone port. All lines use same tone parameters. After modify the tone parameters, you must save modify then Reboot to let the modified parameters work.

Call Forward function: you can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.

When you finished the setting, please click the Submit button.

## Forward Setting

You could set the forward number of your phone in this page.

All Forward: ☐      Busy Forward: ☐      No Answer Forward: ☐

	Name	URL
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

Time Out:  (10~90 sec)

## SNTP settings

This page defines the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

## SNTP Settings

You could set the SNTP servers in this page.

SNTP: ☒ On ☐ Off

Primary Server:	<input type="text" value="192.43.244.18"/>
Secondary Server:	<input type="text" value="208.184.49.9"/>

Time Zone:	GMT	+	<input type="text" value="08"/>	:	<input type="text" value="00"/>	(hh:mm)
Sync. Time:	<input type="text" value="1"/>	:	<input type="text" value="0"/>	:	<input type="text" value="0"/>	(dd:hh:mm)

## Volume Setting

This page defines the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.

Handset Volume is to set the volume for you can hear from the handset.

Ringer Volume is to set the ringer volume for you can hear.

Handset Gain is to set the volume send out to the other side's handset.

## Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	<input type="text" value="10"/>	(0~12)
Ringer Volume:	<input type="text" value="10"/>	(0~10)

Handset Gain:	<input type="text" value="8"/>	(0~15)
---------------	--------------------------------	--------

## Block Setting

This page defines the e Block Setting to keep the phone slience. You can choose Always Block or Block a period.

Always Block: All incoming call will be blocked until disable this feature.

Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

When you finished the setting, please click the Submit button.

## Block Setting

You could set the block period of your phone in this page.

Always Block:	<input type="radio"/> On <input checked="" type="radio"/> Off
Block Period:	<input type="radio"/> On <input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)
To:	<input type="text" value="00"/> : <input type="text" value="00"/> (hh:mm)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

### Caller ID settings

This page defines the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.

## Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:	<input type="text" value="Don't show caller ID"/>
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

### Dial Interval Settings

This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

## Dial Interval Settings

You could set the dial interval in this page.

Dial Interval:	<input type="text" value="5"/> (3~9 sec)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

## Flash Time Setting

When you use the PSTN Phone and you need to press the Hook to do the Flash (Switch to the other phone line or HOLD), this function is for you to set the time you press the Hook to represent the Flash function.

### Flash Time Settings

You could set the flash time in this page.

Flash Time:  (Range:1~200, Unit:10ms)

## Call waiting Settings

When you are talking with other people, You can choose If you want to hear the notice when there is a new coming call. If the call waiting function is On, if there is a new incomeing call, you will hear the call waiting notice in your current call. If you set the function to Off, then you will not hear any notice.

### Call Waiting Settings

You could enable/disable the call waiting setting in this page.

Call Waiting: ☒ On ☐ Off

## DDNS Settings

This page defines the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button.

## DDNS Settings

You could set the configuration of DDNS in this page.

DDNS:	<input type="radio"/> On <input checked="" type="radio"/> Off
Host Name:	<input type="text"/>
User Name:	<input type="text"/>
Password:	<input type="text"/>
E-mail Address:	<input type="text"/>
Type:	<input type="text" value="dyndns"/>
Wild Card:	<input type="text" value="on"/>
BACKMX:	<input type="checkbox"/> On <input type="checkbox"/> Off
Off Line:	<input type="checkbox"/> On <input type="checkbox"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

## Service Domain Settings

This router comes with the built-in firewall based on the advanced technology of Stateful Packet In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the Phon. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

Display Name: you can input the name you want to display.

User Name: you need to input the User Name get from your ISP.

Register Name: you need to input the Register Name get from your ISP.

Register Password: you need to input the Register Password get from your ISP.

Domain Server: you need to input the Domain Server get from your ISP.

Proxy Server: you need to input the Proxy Server get from your ISP.

Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

Register Period: you need to input the Register Period get from your ISP. This is count in minute.

You can see the Register Status in the Status item. If the item shows "Registered", then your Phone Adapter is registered to the ISP, you can make a phone call directly.

If you have more than one SIP account, you can following the steps to register to the other ISP.

When you finished the setting, please click the Submit button.

## Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	<input type="radio"/> On <input checked="" type="radio"/> Off
Display Name:	<input type="text"/>
Line Number:	<input type="text" value="1001"/>
Register Name:	<input type="text"/>
Register Password:	<input type="password"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Register Period:	<input type="text" value="15"/> (0~99) [0: 30 sec,1~99 min]
Status:	Not Registered

## Port Settings

This page defines the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

## Port Settings

You could set the port number in this page.

SIP Port:	<input type="text" value="5060"/>	(1024~65535)
RTP Port:	<input type="text" value="60000"/>	(1024~65535)

## Codec Settings

This page defines the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

# Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.729
Codec Priority 2:	G.723
Codec Priority 3:	G.711 u-law
Codec Priority 4:	G.711 a-law
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

Submit

Reset

## RTP Setting

This page defines the Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

# RTP Settings

You could set the RTP Settings in this page.

Outband DTMF: ☒ On ☐ Off

Send DTMF SIP Info: ☐ On ☒ Off

Submit

Reset

## RPort Settings

This page defines the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

### RPort Settings

You could enable/disable the RPort setting in this page.

RPort: ☐ On ☒ Off

Submit

Reset

## Other Settings

This page defines the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

### Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS:	<input type="text" value="40"/> (0~63)
SIP QoS:	<input type="text" value="40"/> (0~63)
SIP Expire Time:	<input type="text" value="3600"/> (60~86400 sec)

Submit

Reset

## STUN settings

This page defines the STUN Enable/Disable and STUN Server IP address in this page. This function can help your Phone Adapter working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

## STUN Settings

You could set the IP of STUN server in this page.

STUN: ☐ On ☒ Off

STUN Server:   
STUN Port:  (1024~65535)

## Auto Configuration

In Auto Configuration Setting you need to check with your service provider if they have provided this function. Usually this function will be bundle with an IP PBX to use in the office.

### Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

Auto Configuration: ☐ On ☒ Off

## Country Settings

In Country Settings is for you to set the Country, different country will have different settings in FXS interface.

### Country Setting

You could select your country setting in this page.

Select Country:  ▼

## System Authority

In System Authority you can change your login password.

### System Authority

You could change the login username/password in this page.

Username:	<input type="text" value="root"/>
New password:	<input type="password"/>
Confirmed password:	<input type="password"/>

## Save & Reboot

In Save & Reboot you can save the changes you have done. If you want to use new setting in the Phone Adapter, You have to click the Save button. After you click the Save button, the Phone Adapter will automatically restart and the new setting will effect.

### Save & Reboot

You have to save changes to effect them.

Save Changes:

## Firmware Upgrade

In Firmware Upgrade function you can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:

Select the firmware code type, AP or DSP code.

Click the "Browse" button in the right side of the File Location or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the Phone Adapter then click the Update button.

## Firmware Upgrade

You could update the newest firmware.

Code Type: ☒ AP ☐ DSP

File Location:

### Reset to Default

In Default Setting you can restore the Phone Adapter to factory default in this page. You can just click the Restore button, then the Phone Adapter will restore to default and automatically restart again.

## Reset to Default

You could click the restore button to restore the factory settings.

Reset to default:

### Reboot without saving

Reboot function you can restart the Phone Adapter. If you want to restart the Phone Adapter, you can just click the Reboot button, then the Phone Adapter will automatically.

## Reboot without Saving

You could press the reboot button to restart the system.

Reboot without Saving:

## Appendix A Voice communications

There are several ways to make calls to desired destination in VIP-156. In this section, we'll lead you step by step to establish your first voice communication via keypad and web browsers operations.

### Voice communication via IP PBX system – IPX-2000 (Auto-config)

In the following sample, we'll introduce how to integrate the ATA with our IP PBX system IPX-2000 via Auto-config feature.



- VIP-156 IP Address: 192.168.0.1  
Line Number: 1001

- VIP-156 IP Address: 192.168.0.2  
Line Number: 2002

### Machine configurations on the IPX-2000:

#### STEP 1:

Log in IPX-2000 and browse to the DHCP menu and create new options list for the auto configuration.

The screenshot shows the DHCP configuration interface. At the top, there are radio buttons for "Enable" (selected) and "Disable", along with "Save", "Cancel changes", "Delete", and "Show client" buttons. Below this is a "DHCP Pool" section with a blue background and a "<Add new >" link. The "lan" pool is selected. To the right, the "ID" field is set to "lan". The "Single host" checkbox is unchecked. The "Range" is set to "192.168.1.101 ~ 192.168.1.200". The "Options list" section shows a dropdown menu with "lan,150,192.168.1.1" selected. Below this is a table with two columns: "Code" and "Value". The first row has "151" in the "Code" column and "http://192.168.0.50/tftpboot" in the "Value" column. There is an "Add" button next to the "Value" field.

Code: please insert 151 as the DHCP server option.

Value: http://LAN IP of IPX-2000/tftpboot

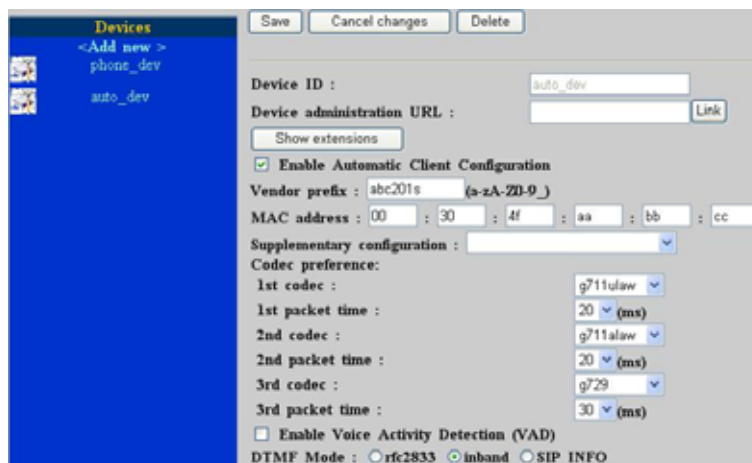
If you'd like to enable auto-config for IP extension features in IPX-2000, please be sure to setup the DHCP option code and the value information.

In most case, insert the optional code 151 and the value=http://192.168.0.50/tftpboot/

**Note:** the 192.168.0.50 is the IP address of IPX-2000

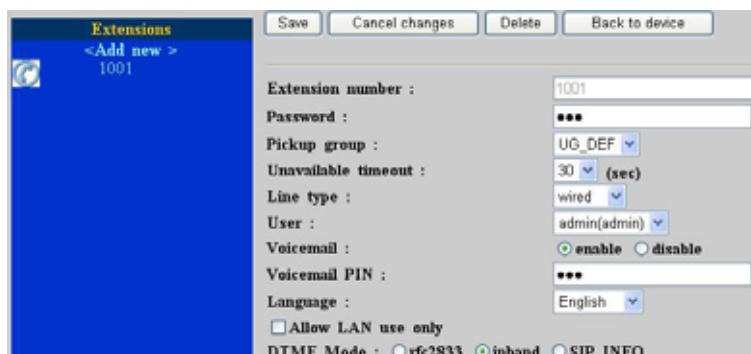
## STEP 2:

Please browse to the Device menu and create new device for the auto configuration.



## STEP 3:

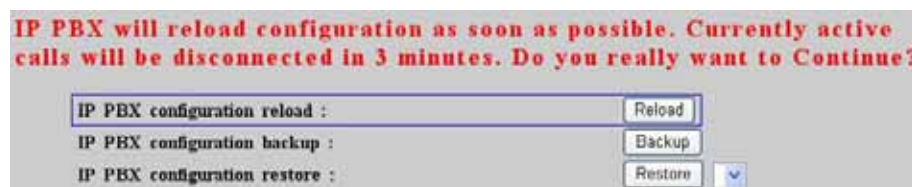
Please press the Show extensions button to create the two extension accounts/password: 1001/123 (for VIP-156 A), and 1002/123(for VIP-156 B) for the voice calls.



## STEP 4:

After setting up the parameters, please refer to the path to activate the settings :**Service --->**

**IP PBX service ---> IP PBX configuration reload**



Machine configurations on the VIP-156:

## STEP 5:

Please log in VIP-156 via web browser, browse to the **Advanced Settings** menu. In the setting page, please browse to the Auto-config page, and enable the Auto Configuration features for IP PBX system.

## Auto Configuration Setting

---

You could enable/disable the auto configuration setting in this page.

Auto Configuration: ☒ On ☐ Off

### STEP 6:

After enabling the Auto-config feature, the VIP-156 shall be able to obtain IP address and SIP extension information from IP PBX system IPX-2000 information. To verify the auto-config results, you may connect telephone set to VIP-156, press **#120#** to check if the IP address is obtained from IPX-2000. And **#122#** can be used to verify the extension number assigned by IPX-2000.

### STEP 7

Repeat the same configuration steps on VIP-156 B, and check if the VIP-156 is successfully registered with the IPX-2000 as one of the IP extensions.

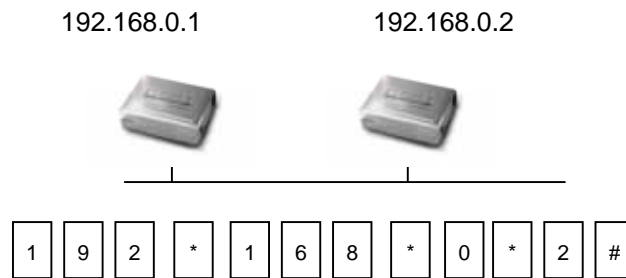
### Test the scenario:

To verify the VoIP communication, you may make calls from extension side (VIP-156 A) 1001 to the number 1002 (VIP-156 B) or reversely make calls from extension client (VIP-156 B) 1002 to the number 1001 (VIP-156 A)

## VIP-156 to VIP-156 connection via IP address

Assume there are two VIP-156's in the network the IP address are 192.168.0.1, 192.168.0.2

Analog telephone sets are connected to the **phone** (RJ-11) port of VIP-156s respectively



### Operation steps:

Pick up the telephone set on VIP-156 A, you should be able to hear the dialtone, press the keypad: 192\*168\*0\*2# shall be able to connect to the VIP-156 B.

Then the phone in 192.168.0.2 should ring. Please repeat the same dialing steps on VIP-156 B to establish the first voice communication from the second VIP-156

### Hint

- If the IP address of the remote calling party is known, you may directly make calls via its IP address and end with an "#".
- If the IP phones are installed behind a NAT/firewall/IP sharing device, please make sure the NAT device support SIP applications before making calls

## Voice communication via SIP proxy server –SIP50



■ VIP-156 IP Address: 192.168.0.1  
Line Number: 1001

■ VIP-156 IP Address: 192.168.0.2  
Line Number: 2002

### Machine configurations on the VIP-156:

#### STEP 1:

Log in SIP-50 and create two testing accounts/password: 1001/123 (for VIP-156 A), and 1002/123(for VIP-156 B) for the voice calls.

## STEP 2:

Please log in VIP-156 via web browser, browse to the **SIP setting** menu and select the **Domain Service** config menu. In the setting page, please insert the account/password information obtained from your service provider (in this sample, we're using PLANET SIP-50 as the SIP Proxy server for SIP account, call authentications), and then the sample configuration screen is shown below:

### Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	<input type="text" value="1001"/>
Line Number:	<input type="text" value="1001"/>
Register Name:	<input type="text" value="planet"/>
Register Password:	<input type="password" value="•••"/>
Domain Server:	<input type="text" value="192.168.0.50"/>
Proxy Server:	<input type="text" value="192.168.0.50"/>
Outbound Proxy:	<input type="text"/>
Register Period:	<input type="text" value="15"/> (0~99) [0: 30 sec, 1~99 min]
Status:	Registered

## STEP 3:

Repeat the same configuration steps on VIP-156 B, and check the machine registration status, make sure the registrations are completed.

### Test the scenario:

To verify the VoIP communication, you may make calls from SIP client (VIP-156) 1001 to the number 1002 (VIP-156) or reversely make calls from SIP client (VIP-156) 1002 to the number 1001 (VIP-156 A)

## Appendix B VIP-156 / VIP-156PE Specifications

Product	SIP Telephone Adapter	
Model	VIP-156	VIP-156PE
Hardware		
LAN	1 x 10/100Mbps RJ-45 port (802.3af PoE for VIP-156PE)	
PC	1 x 10/100Mbps RJ-45 port	
PHONE	1 x RJ-11 connection	
Protocols and Standard		
Standard	SIP 2.0 (RFC3261)	
Voice codec	G.723.1 (6.3k/5.3k), G.729A, G.729B, G.711	
Voice Standard	Voice activity detection (VAD) Comfort noise generation (CNG) Acoustic echo canceller (AEC) G.165: Line echo canceller (LEC) Jitter Buffer	
Protocols	SIP 2.0 (RFC-3261), TCP//IP, UDP/RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, PPP, PPPoE	
Network and Configuration		
Access Mode	Static IP, PPPoE, DHCP	
Management	Web, keypad	
Dimension (W x D x H)	94 x 72 x 30 mm	
Operating Environment	0~40 degree C, 10~95% humidity	
Power Requirement	12V DC	
EMC/EMI	CE, FCC Class B	